MULTIBAND DIGITAL GAIN CONTROLLER

J. C. Ventura
Laboratoire d’Electromagnétisme et Acoustique
Ecole Polytechnique Fédérale de Lausanne
EPFL - 1015 Lausanne, Switzerland

ABSTRACT

The restoring of correct loudness functions for hearing impaired requires input/output functions which depend on both, amplitude and frequency. The described gain controller does so by splitting the input spectrum in three bands and correcting the loudness functions in each separately. The input/output static characteristics are fully programmable and the gain is piloted by an envelope detection based on speech properties.

1. INTRODUCTION

Hearing Impairment is often characterized by a loss of the listening dynamic range. Not only the hearing threshold rises but also the subjective loudness perception is altered. Moreover this deformation is frequency dependant.

A first attempt to improve speech intelligibility is done by adapting the speech loudness differences to the patient’s individual loudness perception. This is done by splitting the speech signal into three frequency bands, the gain in each being contiguously varied according to a preprogrammed input/output level static characteristic (SC). This is the task of the Programmed Gain Controller (PGC).

The SC is graphically programmed by segments according to the patient’s psychoacoustical data (loudness category scaling).

It is also important to achieve a correct transient behavior which is dependant on the effectiveness of the input level detection.

2. CHANNEL SPLITTING

Channel splitting into several bands is a means to deal with the frequency dependance of the pathology. A loudness restoration gain correction is approached within a band in which the ear characteristics (relative to a normal ear) vary slightly. Therefore the ideal number of channels varies among different subjects. On the other hand, by increasing the number of bands, one increases the number of possible discontinuities in the crossover regions, not to mention the associated difficulties in hardware requirements.

This work’s experiments are carried out in a three band configuration, the crossover frequencies being programmable.

The implanted filters are butterworth IIR with 24dB/oct slopes. An example is shown in fig.1 (1):

i) lowpass, bandpass and highpass filter characteristics are shown overlapped. The crossover frequencies are marked by the dotted lines.

ii) Output characteristic of the three summed channels.

3. GAIN CONTROLLERS

3.1. BLOCK DIAGRAM

The PGC operates as shown in Fig.2: the detected input level value (envelope detector) is used to address a lookup table which yields the corresponding gain. The output is the product of the gain and the input signal. The latter is delayed in order to compensate the level detection intrinsic delay.
3.2. STATIC CHARACTERISTIC

Since the SC is stored in a lookup table it can have any desired shape, depending on the editing facilities allowed to the user. In this realization he can graphically determine the SC diagram by placing the kneepoints of a broken line that best approach the desired curve (Fig. 3).

![Fig. 3](image)

3.3. DYNAMIC BEHAVIOR

It is important that the level detector is fast enough to faithfully capture the signal transients. Meanwhile it must be insensitive to the waveform instant amplitude variations. A slow detector causes overshoots at the occurrence of short attack times. Percussive loud noises such as object crashes or door slams will be amplified rather than attenuated. A slow decay attenuates the low levels immediate to higher ones. This effect causes the masking of consonants. On the other hand, waveforms would be deformed by a detector fast enough to capture their instant variations, thus causing distortion.

The envelope extraction by nonlinear filtering is the chosen method for level detection. It is based on a model which considers speech as the product of a rapidly varying carrier and a slowly varying envelope [1]. By taking the signal logarithm this product is turned into an addition. The slow and fast components will then appear at different frequency regions. Although these two are not perfectly parted, the envelope has most of its energy within the band below 20Hz. The rest of the spectrum has its energy spread over a wide high frequency band.

Fig. 4 illustrates the simulation of the successive steps taken to extract the envelope of the french words Le clavier. Specters of the successive signals are displayed at the right:

i) sampled original signal.

ii) shows its logarithm, calculated after having had the signal rectified.

iii) logarithm of the envelope after low-pass filtering. This signal commands the instant gain.

![Fig. 4 i)](image)

![Fig. 4 ii)](image)

![Fig. 4 iii)](image)

3.4. IMPLEMENTATION

An algorithm performing the described processing has been implemented in a TMS32010 processor. The SC is graphically edited in a personal computer which calculates the gain lookup table. Between two kneepoints the gain \((g)\) is calculated as follows:

\[
g = 10^{G/20}
\]

\[
G = (1/R-1)\text{Lin} + \text{Lout 1} - \text{Lin 1}/R ;
\]

\[
R = (\text{Lin 2} - \text{Lin 1})/(\text{Lout 2} - \text{Lout 1}) ;
\]

\[
\text{Lin} = \text{input level} ;
\]

Lin 1, Lout 1, Lin 2, Lout 2 are respectively the input and output levels for two successive kneepoints.

The detection branch works as shown in Fig. 5. The 12 bit input is rectified, coded in an 8 bit log scale and low pass filtered. The starting address in the lookup table is then added to the filter output, yielding the address of the instant gain. The filter is an IIR 2nd order Butterworth with a 20Hz cutoff frequency. The delay to be compensated...
is approximately 11.5 ms.

Fig. 5

Fig. 6 shows the processing steps over sampled speech (french words Le clavier) as they are processed by the TMS:

i) original signal;
ii) detected envelope (in a log scale, corresponding to step iii) in fig.3;
iii) output signal.

The programmed SC was a compressor of 4:1 ratio.

Fig. 6

Fig. 7 illustrates the compressor reaction to a percussive sound occurred during speech processing (bump on the microphone in the middle of the word bonsoir).

3.5. ACCURACY

The input/output signal is 12 bit coded. The Lin/Log conversion codes the 2048 possible input values (unsigned) in 8 bit words, approaching the log characteristic by segments (fig. 8). The maximum error compared to a true log characteristic is 0.44 dB (fig. 9).

Fig. 8

Fig. 9

Figures 10 and 11 show measures of static characteristic (corresponding to programming on fig. 2) and harmonic distortion.
4. APPLICATION ENVIRONMENT

The described PGC is used in a multiprocessor array system. It contains six processors controlled by a personal computer [4]. Each processor's task is either a band split filter or a PGC (Fig. 12). It is aimed to perform multiband gain control (3 bands), thus approaching a frequency dependant correction of the pathological loudness perception.

The SCs are programmed according to the patients audiometric data. They are determined by the measured Magnitude Loudness Estimation curves [2], [3].

5. IMPROVEMENTS

The recent developments of digital signal processors concerning accuracy (bit resolution), speed and programming facilities allows the implant of the described algorithms in a single chip while improving the overall performances. Dynamic range is improved due to a better bit resolution (word lengths are larger in both data memory and accumulators).

6. CONCLUSION

The development of programmable gain controllers using digital techniques can assure the best accuracy in determining the input/output characteristic while showing a good transient response. Its programming is also eased.

Although the level detection method is based on speech properties, its behavior regarding interfering non-speech sounds is most satisfactory. Moreover this principle can be used within analog hearing aids for which multiband processing is beginning to break through.

7. ACKNOWLEDGEMENTS

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8. REFERENCES